IN THE SPECIFICATION

Please amend the last paragraph on page 6 as follows:

Referring now to the drawings, wherein like reference numerals designate identical or corresponding parts throughout the several views, Figures 1 illustrates a communications system 100 including a data packet network 110 such as the Internet. Connected to the data packet network are end user terminal 120 and end user terminal 130, which is connected to the data packet network 110 through the PSTN (public switched telephone network) 171 using a typical dial-up connection to an Internet service provider with a modem (not shown in Figure 1). End users at the terminals 120, 130 may view web pages from a variety of sources, including merchant web servers ~~140,142~~ 141, 143 which are also connected to the data packet network 110.

Please amend the first full paragraph on page 7 as follows:

Each of the merchant web servers includes an icon for establishing a VoIP telephone call to an associated call center. Preferably, the icon causes an Internet telephony software application, which may be in the form of an applet, to be downloaded to an end user terminal 120, 130 and automatically establish the telephone call without requiring previously-installed software at the end user terminal 120, 130. When the web page and icon are provided by the merchant web server 141, a "pure" internet telephony call is established between the respective end user terminal 120, 130 and the VoIP-ready call center 150 associated with the merchant web server 141 (as indicated by dashed line 142). In this case, packets are exchanged between the respective end user terminal 120, 130 and the call center 150 directly through the data packet network 110 without using the PSTN (except for the portion of the PSTN 171 through which the end user terminal [[132]] 130 is connected to the data packet network 110).

Please amend the second full paragraph on page 7 as follows:

When the web page [[an]] <u>and</u> icon are provided by merchant web server 143, the VoIP telephone call must be routed through a gateway 160 and the PSTN 172 to a conventional call center 180, which is not VoIP-ready, associated with the merchant web server 143 as indicated by the dashed line 144. In this situation, a packet is sent from an end user terminal 120, 130 to the gateway 160. The gateway 160 unpacks the packet, converts the digital information to analog form, and transmits it to the call center 180 over the PSTN 172 (PSTN 171 and PSTN 172 may be part of the same public switched telephone network, but are shown separately in Figure 1 for the purposes of illustration). The gateway also receives voice information in analog form from the PSTN call center 180, digitizes and packetizes the analog information, and sends the packets to the appropriate end user terminal 120, 130.